REMARKS/ARGUMENT

Request for Personal Interview:

Attached hereto is a Form PTOL-413A. Applicants' representative will contact the Examiner by telephone to set a mutually convenient date and time.

Regarding the Claims in General:

Claims 2-8, 10-22, 27, 29-55 and 60-61 are now pending. Claims 4, 16, and 60 have been amended to clarify the original intent of one of the limitations thereof without narrowing the scope of the claim. Minor formal corrections corrections have also been made in claims 4, 31, 32, 50, 55, and 60. Claim 9 has been canceled without prejudice and is replaced by claim 61 which is of the same scope, but expresses the claimed concept more clearly in terms of the underlying mathematical relationship, rather than in words.

Regarding the Prior Art Rejections:

In the outstanding Office Action, claims 2-17, 30-53, and 60 were rejected as unpatentable over Adlersberg et al. U.S. Patent 5,012,519 (Adlersberg) in view of Cooper U.S. Patent 5,790,671(Cooper). Reconsideration and withdrawal of this rejection is respectfully requested.

Preliminarily, the Examiner's attention is respectfully directed to the fact that nowhere in Section 2 of the Office Action is there any explanation as to the role of Cooper in this rejection. Indeed, Cooper is not discussed, or even mentioned except in the statement of the rejection. Thus, it is not seen how this rejection can be properly responded to.

Nevertheless, Adlersberg differs from the present invention in numerous ways, and there is nothing apparent in Cooper which overcomes these deficiencies.

Most obviously, the present invention is directed to compression of an audio signal, the noise and information components of which are to be stored in digital form, whereby the storage capacity of the storage medium can be minimized. Adlersberg, on the other hand, and also Cooper, are directed to *noise reduction* for signals in communication channels. Both references function by

isolating and reducing the noise components of incoming signals on a continuing basis. System data storage capacity is not an issue in either instance.

More particularly, Adlersberg's method for reducing the noise in a noisy speech signal involves subjecting the speech signal as a continuous data stream to a fast-Fourier-transformation, and processing the transformed signal *in the frequency domain* by a noise reduction stage. The processed signal is then transformed back into the time domain, so that an "enhanced speech signal" is obtained (column 5, lines 28 - 43). Adlersberg does not perform any compression of the (noisy) speech signal in the time domain.

On the other hand, the present invention functions by periodically obtaining short duration samples of the audio signal, normalizing the samples (or optionally first filtering, band limiting or digitizing) to a predetermined range of values or amplitudes D, and subsequently non-linearly transforming (mapping) the normalized signals to a second amplitude range W, and then storing the resulting signals in an electronic memory for later use. The range W may be smaller or equal to D, but it is preferably substantially smaller. The non-linear transformation serves the purpose of amplifying predetermined desired areas of range D to enhance the intelligibility of the resulting stored signal. The selective amplification is based on predetermined parameters, and is not adaptive in the sense of the dynamic frequency domain processing as shown in Adlersberg. It is also purely a time domain, and not a frequency domain process, in any event.

These differences are clearly reflected in the claims now before the Examiner. For example, claim 60 is directed to a method for storing an electric signal representing recorded ambient noise in compressed form. The first step of the claimed method is:

> periodically recording samples of the ambient noise using a sound transducer, the sample duration being shorter than the sampling cycle

Adlersberg can not do this. The only analogy to the claimed samples in Adlersberg concerns breaking the continuous data stream corresponding to the incoming speech signal into a sequence of overlapping frames or windows (see column 4, line 65; column 4, lines 57 - 65) as a precursor to performance of the FFT and is part of the A to D conversion process. To emphasize this difference, but without charging the scope thereof, claim 60 has been amended to specify that the sample

duration is shorter than the sampling cycle. In Adlersberg, the duration of the overlapping frames is obviously not shorter than the frame rate repetition period. Also, described at column 12, beginning at line 13, the A to D conversion is performed continuously in real-time, and a contiguous, non-interrupted sequence of samples at a rate of 8,000 samples per second is generated (column 12, line 25).

Claim 60 further calls for the step of:

> normalizing the amplitude of a signal output of the transducer or a signal derived therefrom within a first predetermined range D;

The initial AGC in Adlersberg does not meet this recitation. For one thing, the initial AGC is controlled so that "no switching side effects are heard at the digital processing system output", i.e., in the final data stream before perhaps digital-analog-conversion back to an audio signal (column 5, lines 11 -13). Furthermore, the control parameters for the AGC are calculated. based on signal energy, not on signal amplitude (column 4, lines 53-54). As described in the reference, the signal energy which is related to the square of the amplitude, is obtained by summering up over the time covered by the respective frame, i.e. by a brute force approach, with the mean square value of the amplitude taken as the reference value. Accordingly, large amplitudes have more weight than small amplitudes due to squaring, and as a consequence of using a mean value, it is sure that some (energy) values are even higher than the reference values. By normalization over a range D, a suitable maximal signal amplitude (i.e., D) can be selected to avoid distortion due to clipping. Adlersberg does not contemplate this.

The next recited step is:

> mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values using a non-linear mapping function to obtain an emphasis of selected values ranges within the first or the second predetermined ranges.

In Adlersberg, there is no mapping to a second amplitude range of a normalized signal sample using a non-linear function: column 7, line 10 et seq. of the reference pertains to treating the signal in the frequency domain, i.e. an FFT has already been performed applied. Time domain, i.e., amplitude processing is not involved.

The "amplitude estimators" mentioned in column 7, line 43 are not comparable to the compressed sample values obtained by the present invention: These are estimates of the Fourier coefficients of the speech portions in the noisy signal, and are used to recover the incoming speech signal with a reduced noise level. Hence, these estimators are neither practically nor theoretically comparable to normalized amplitude values mapped non-linearly to a second range of values. The Examiner's interpretation of the claim language and of the reference is entirely too broad.

Finally, claim 60 calls for "storing the mapped result in an electronic memory in a digital format". Adlersberg obviously does not store non-linearly amplitude-mapped sample signals either as a final objective, or as an intermediary stage.

Although the Examiner has not indicated what deficiencies in Adlersberg he proposes overcome by reference to Cooper, it is clear that none of the features of claim 60 are found in that reference either.

Independent method claim 4 is similar to claim 60, and is distinguishable over Adlersberg and Cooper for the reasons stated above. In addition, this claim calls for:

> dividing the audio signal into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and wherein any content of the other band signals contained in each band signal is present only in an attenuated form.

Adlersberg's Fourier transformation is obviously not a time domain process, in contrast to the time domain process involved in "direct" i.e., audio signal filtering. To emphasize this difference, claim 4 has been rearranged to place the filtering step before the normalizing step, and other conforming changes have also been made. It should be noted, however, that this rearrangement is not intended to, and does not change the scope of the claim.

Independent method claim 16 is also similar to claim 60, and is distinguishable over Adlersberg and Cooper for the reasons stated above. In addition, this claim specifies that the range of normalized values D is defined by a lower limit D_u, and an upper limit D_o, and the normalization is effected by:

> obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or

> derived signal, which is shorter than or equal to the duration of a hearing sample,
>
> multiplying the reciprocal value of said maximum by (D_o - D_u +1), and
>
> multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

There is nothing like this anywhere in Adlersberg or Cooper, and certainly not in the portion of Adlersberg cited at the top of page 4 of the Office Action.

For the foregoing reasons, claims 4, 16, and 60 are allowable.

Claims 2, 3, 11-15, 17-22, 27, 29-32, and 42-48 are dependent on allowable claim 60, claims 5-10, and 33-41 are dependent on allowable claim 4, and claims 49-53 are dependent on allowable claim 16. These claims are allowable for the reasons stated above. Irrespective of whether it would be obvious, for example to use Adlersberg's noise reduction system in a broadcast program monitoring system such as Kenyon et al., nothing in Kenyon or any of the other secondary references overcomes the basic deficiencies in Adlersberg.

In addition, the indicated dependent claims recite features which, in combination with the features of their respective parent claims are neither taught nor suggested in the references, either alone or in combination.

In view of the foregoing, favorable reconsideration and allowance of this application are respectfully solicited.

I hereby certify that this correspondence is being transmitted via facsimile (703) 872-9306 on February 23, 2004:

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Respectfully submitted,

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